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EXAMINER	
SIEDLER, DOROTHY S	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

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Office Action Summary	Application No. 10/685,585	Applicant(s) LIU ET AL.	
	Examiner Dorothy Sarah Siedler	Art Unit 2626	

-- *The MAILING DATE of this communication appears on the cover sheet with the correspondence address* --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 05 December 2007.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☐ Claim(s) _____ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-12, 14-31 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

Applicant's arguments, see Arguments/Remarks, filed December 5, 2007, with respect to the rejection(s) of claim(s) 1-31 under 35 U.S.C. §103 have been fully considered and are persuasive. Therefore, the rejection has been withdrawn. However, upon further consideration, a new ground(s) of rejection is made in view of ***Colbath*** "Spoken Document: Creating Searchable Archives from Continuous Audio" IEEE 2000).

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claims 1-12 and 14-31 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***Colbath*** ("Spoken Document: Creating Searchable Archives from Continuous Audio" IEE 2000) in view of ***Liu*** ("Fast Speaker Change Detection for Broadcast News Transcription and Indexing", and further in view of ***Stanford*** (5,475,792).

As per claim 1, ***Colbath*** discloses a method for classifying an audio signal containing speech information, the method comprising:

Receiving the audio signal (page 2, Component Technologies, *audio wave file*);

Classifying the sound in the audio signal based on at least one non-phoneme based model (page 3, Speaker Segmentation, Clustering, and Identification, *the speakers are classified by gender for segmentation and identification*).

Colbath does not disclose classifying a sound in the audio signal as a vowel class when a first phoneme-based model determines that the sound corresponds to a sound represented by a set of phonemes that define vowels, classifying the sound in the audio signal as a fricative class when a second phoneme-based model determines that the sound corresponds to a sound represented by a set of phonemes that define consonants, and classifying the sound in the audio signal based on at least one non-phoneme based model, the at least one non-phoneme model including at least one model for classifying the sound in the audio signal based on bandwidth. However, ***Colbath*** does disclose the use of a speech recognition component and a speaker segmentation component as part of a system designed to transform an audio wave file into an indexed database (page 2, Component Technologies), but does not provide further details on either component. ***Liu*** discloses classifying a sound in the audio signal as a vowel class when a first phoneme-based model determines that the sound corresponds to a sound represented by a set of phonemes that define vowels (section 3, Phone-Class Decode and Figure 1), classifying the sound in the audio signal as a

fricative class when a second phoneme-based model determines that the sound corresponds to a sound represented by a set of phonemes that define consonants (section 3, Phone-class decode and Figure 1). ***Liu*** discloses a fast speaker change detection algorithm for fast transcription and audio indexing of spoken broadcast news (Abstract).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use the phone-based models of ***Liu*** in ***Colbath***, since it would result in improvements in speaker change detection accuracy, speech recognition accuracy and speed, as disclosed in ***Liu*** (Abstract).

Additionally, ***Stanford*** discloses a speech recognition system that enables recognition of high bandwidth or telephony (low bandwidth) speech signals (column 2 lines 30-32 and column 8 lines 36-44). ***Stanford*** states that low bandwidth speech reduces the accuracy of speech recognizers (column 3 lines 37-39), and discloses a system that trains and uses two separate codebook and phoneme models, one for low bandwidth speech and one for high bandwidth speech (column 8 lines 36-44). The addition of the low bandwidth recognition model improves recognition accuracy for low bandwidth input, such as telephone speech.

Therefore it would have been obvious to one or ordinary skill in the art at the time of the invention to classifying the sound in the audio signal based on at least one non-phoneme based model, the at least one non-phoneme model including at least one model for classifying the sound in the audio signal based on bandwidth in ***Colbath***,

since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp in order to achieve the predictable result of accurately recognizing and segmenting audio information for indexing, regardless of the input quality.

As per claim 2, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 1, and ***Colbath*** further discloses wherein the at least one non-phoneme based model includes models for classifying the sound in the audio signal based speaker gender (page 3, Speaker Segmentation, Clustering, and Identification, *the speakers are classified by gender for segmentation and identification*).

As per claim 3, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 1, however ***Colbath*** does not disclose wherein the at least one non-phoneme based model includes a model for classifying the sound in the audio signal as silence. ***Liu*** discloses wherein the at least one non-phoneme based model includes a model for classifying the sound in the audio signal as silence (Section 3 Phone-Class Decode, paragraph 3, *silence model*).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use a silence model of ***Liu*** as a non-phoneme based model in ***Colbath***, since it is a much more effective model than energy based models for detecting non-speech regions, as indicated in ***Liu*** (page 3, section 3 Phone-Class

Decode, paragraph2), which in turn improves speaker change detection and speaker clustering or identification, as indicated in ***Liu*** (page 3, section 3, Phone-Class Decode).

As per claims 4 and 5, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 1, but ***Colbath*** does not disclose initially converting the audio signal into a frequency domain signal, and generating cepstral features for the audio signal. However, during standard speech processing, input speech is converted into a frequency domain representation by Fourier Transform, then converted into specific feature vectors, such as cepstral vectors. This is confirmed by ***Liu***, which discloses the use of cepstral vectors for speaker change detection (section 4 Speaker Change Detection).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to convert the audio signal to a frequency domain signal, and generate cepstral features in ***Colbath***, since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp in order to achieve the predictable result of determining reliable feature vectors for speech processing.

As per claim 6, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 1, however ***Colbath*** does not disclose wherein the fricative class includes phonemes that relate to fricatives and obstruents. ***Liu*** discloses wherein the fricative class includes phonemes that relate to fricatives and obstruents (Section 3

Phone-Class Decode, paragraphs 3 and 4, *fricatives are a specific type of obstruent, therefore it is inherent that the class includes phonemes that relate to obstruents*).

Therefore it would be obvious to one of ordinary skill in the art at the time of the invention to have a fricative class that includes phonemes that relate to fricatives and obstruents in ***Colbath***, since it would result in improvements in speaker change detection accuracy, speech recognition accuracy and speed, as disclosed in ***Liu*** (Abstract).

As per claim 7, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 1, however ***Colbath*** does not disclose wherein the first and second phoneme-based models are Hidden Markov Models. However, Hidden Markov Models are statistical models commonly used as phoneme models in speech and language processing. In addition, ***Liu*** discloses wherein the first and second phoneme-based models are Hidden Markov Models (Section 3 Phone-Class Decode, paragraph 6).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use HMM for the phone-based models in ***Colbath***, since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp in order to achieve the predictable result of determining robust and reliable phone-based models.

As per claim 8, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 1, however ***Colbath*** does not disclose classifying the sound in the audio signal as a coughing class when the sound corresponds to a non-speech sound. ***Liu*** does not explicitly disclose classifying the sound in the audio signal as a coughing class when the sound corresponds to a non-speech sound, however ***Liu*** does disclose coughing as a common non-speech sound (Section 2 Evaluation Metrics, second paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to classify the sound in the audio signal as a coughing class when the sound corresponds to a non-speech sound in ***Colbath***, since classification of coughing as non-speech would enable the exclusion of those frames during speaker clustering for identifying speakers, as taught by ***Liu*** (Section 3 Phone-Class Decode, first paragraph), thus improving speaker segmentation.

As per claim 9, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 1, however ***Colbath*** does not disclose wherein the non-speech sound includes at least one of coughing, laughter, breath, and lip-smack. ***Liu*** discloses wherein the non-speech sound includes at least one of coughing, laughter, breath, and lip-smack (Section 2 Evaluation Metrics, second paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to identify non-speech sounds as coughing, laughter, breath and lip-

smack in ***Colbath***, since classification of non-speech frames enables the exclusion of those frames during speaker clustering for identifying speakers, as taught by ***Liu*** (Section 3 Phone-Class Decode, first paragraph), which improves speaker segmentation.

As per claim 10, ***Colbath*** discloses a method of training audio classification models, the method comprising:

Receiving a training audio signal (page 7-8, Annotation, *training data*);

Receiving phoneme classes corresponding to the training audio signal (page 7-8, Annotation, *training data*);

However, ***Colbath*** does not disclose training a first Hidden Markov Model (HMM), based on the training audio signal and the phoneme classes, to classify speech as belonging to a vowel class when the first HMM determines that the speech corresponds to a sound represented by a set of phonemes that define vowels, training a second HMM, based on the training audio signal and the phoneme classes, to classify speech as belonging to a fricative class when the second HMM determines that the speech corresponds to a sound represented by a set of phonemes that define consonants, and training at least one model to classify the sound based on a bandwidth of the sound. ***Liu*** discloses training a first Hidden Markov Model (HMM), based on the training audio signal and the phoneme classes, to classify speech as belonging to a vowel class when the first HMM determines that the speech corresponds to a sound represented by a set

of phonemes that define vowels (Section 3 Phone-Class Decode, paragraphs 3 and 4), training a second HMM, based on the training audio signal and the phoneme classes, to classify speech as belonging to a fricative class when the second HMM determines that the speech corresponds to a sound represented by a set of phonemes that define consonants (Section 3 Phone-Class Decode, paragraphs 3 and 4).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use the phone-based models of ***Liu*** in ***Colbath***, since it would result in improvements in speaker change detection accuracy, speech recognition accuracy and speed, as disclosed in ***Liu*** (Abstract).

Additionally, ***Stanford*** discloses a speech recognition system that enables recognition of high bandwidth or telephony (low bandwidth) speech signals (column 2 lines 30-32 and column 8 lines 36-44). ***Stanford*** states that low bandwidth speech reduces the accuracy of speech recognizers (column 3 lines 37-39), and discloses a system that trains and uses two separate codebook and phoneme models, one for low bandwidth speech and one for high bandwidth speech (column 8 lines 36-44). The addition of the low bandwidth recognition model improves recognition accuracy for low bandwidth input, such as telephone speech.

Therefore it would have been obvious to one or ordinary skill in the art at the time of the invention to train at least one model to classify the sound based on a bandwidth of the sound in ***Colbath***, since one of ordinary skill in the art has good reason to pursue

the options within his or her technical grasp in order to accurately recognize and segment audio information for indexing, regardless of the input quality of the audio.

As per claim 11, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 10, however ***Colbath*** does not discloses wherein the phoneme classes include information that defines word boundaries. ***Liu*** further discloses wherein the phoneme classes include information that defines word boundaries (Section 5 Experiments and Results, Word-Error-Rate (WER), *the system determines the word error rate, or word recognition accuracy. Therefore it is inherent that the phoneme classes include information on word boundaries*). ***Liu*** also discloses that, for speaker change detection, it is important that speaker boundaries are not labeled in the middle of words (section 3 Phone-Class Decode).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to include information about word boundaries in the phoneme classes in ***Colbath***, sine it would increase speaker segmentation accuracy, as indicated in ***Liu*** (section 3 Phone-Class Decode).

As per claim 12, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 11, however ***Colbath*** does not discloses receiving a sequence of transcribed words corresponding to the audio signal, and generating the information that defines the word boundaries based on the transcribed words. ***Liu*** discloses receiving a sequence of transcribed words corresponding to the audio signal (Section 2 Evaluation

Metrics, last paragraph, *reference transcription*), and generating the information that defines the word boundaries based on the transcribed words (Section 2 Evaluation Metrics, last paragraph, *the reference transcription is aligned with the acoustic data*).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to transcribe words corresponding to the audio signal, and generate the information that defines the word boundaries based on the transcribed words in ***Colbath,*** since the alignment creates a more reliable ground truth for the evaluation of errors, as indicated in ***Liu*** (section 2 Evaluation Metrics).

As per claim 14, this claim recites limitations similar to those recited in claim 2, and is therefore rejected for similar reasons.

As per claim 15, this claim recites limitations similar to those recited in claim 6, and is therefore rejected fro similar reasons.

As per claim 16, ***Colbath*** discloses audio classification device comprising:

A decoder configured to classify portions of the audio signal as belonging to at least one of a plurality of classes (page 3, Speaker Segmentation, Clustering, and Identification, *the speakers are classified by gender for segmentation and identification*).

However, ***Colbath*** does not disclose a signal analysis component configured to receive an audio signal and process the audio signal by at least one of converting the audio signal to the frequency domain and generating cepstral features for the audio signal, and wherein the classes include a first phoneme-based class that applies to the audio signal when a portion of the audio signal corresponds to a sound, represented by a set of phonemes that define vowels, a second phoneme-based class that applies to the audio signal when a portion of the audio signal corresponds to a sound represented by a set of phonemes that define consonants, and at least one non-phoneme class, wherein the decoder determines the at lest one non-phoneme class models that classify the portions of the audio signal based on bandwidth. However, during standard speech processing, input speech is converted into a frequency domain representation by Fourier Transform, then converted into specific feature vectors, such as cepstral vectors. This is confirmed by ***Liu***, which discloses the use of cepstral vectors for speaker change detection (section 4 Speaker Change Detection). ***Liu*** also discloses classes that include a first phoneme-based class that applies to the audio signal when a portion of the audio signal corresponds to a sound, represented by a set of phonemes that define vowels, and a second phoneme-based class that applies to the audio signal when a portion of the audio signal corresponds to a sound represented by a set of phonemes that define consonants (Section 3 Phone-Class Decode, paragraphs 3 and 4).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to convert the audio signal to a frequency domain signal and generate

cepstral features in ***Colbath***, since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp in order to achieve the predictable result of determining reliable feature vectors for speech processing. In addition, the use of a first and second phoneme-based model would result in improvements in speaker change detection accuracy, speech recognition accuracy and speed, as disclosed in ***Liu*** (Abstract).

Additionally, ***Stanford*** discloses a speech recognition system that enables recognition of high bandwidth or telephony (low bandwidth) speech signals (column 2 lines 30-32 and column 8 lines 36-44). ***Stanford*** states that low bandwidth speech reduces the accuracy of speech recognizers (column 3 lines 37-39), and discloses a system that trains and uses two separate codebook and phoneme models, one for low bandwidth speech and one for high bandwidth speech (column 8 lines 36-44). The addition of the low bandwidth recognition model improves recognition accuracy for low bandwidth input, such as telephone speech.

Therefore it would have been obvious to one or ordinary skill in the art at the time of the invention to have the decoder determine the at least one non-phoneme class using models that classify the portions of the audio signal based on bandwidth in ***Colbath***, since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp in order to accurately recognize and segment audio information for indexing, regardless of the input quality of the audio.

As per claim 17, this claim recites limitations similar to those recited in claim 2, and is therefore rejected for similar reasons.

As per claim 18, this claim recites limitations similar to those recited in claim 7, and is therefore rejected for similar reasons.

As per claim 19, this claim recites limitations similar to those recited in claim 2, and is therefore rejected for similar reasons.

As per claim 20, this claim recites limitations similar to those recited in claim 3, and is therefore rejected for similar reasons.

As per claim 21, this claim recites limitations similar to those recited in claim 8, and is therefore rejected for similar reasons.

As per claim 22, this claim recites limitations similar to those recited in claim 9, and is therefore rejected for similar reasons.

As per claim 23, ***Colbath*** discloses a system comprising:

An indexer configured to receive input audio data and generate a rich transcript from the audio data (page 2, second column, *Roungh'n'Ready audio indexing system*) the indexer including:

Audio classification logic configured to classify the input audio data into at least one of a plurality of broad audio classes, the broad audio classes including a non-phoneme based gender class (page 3 Speaker Segmentation, Clustering, and Identification, *the speakers are classified by gender for segmentation and identification*);

A speech recognition component configured to generate the rich transcription based on the broad audio classes determined by the audio classification logic (page 2, Component Technologies, first paragraph);

A memory system for storing the rich transcription; and a server configured to receive requests for documents and respond to the requests by transmitting one or more of the rich transcriptions that match the requests (page 4-5, System Architecture, server and browser); and

A server configured to receive requests for documents and respond to the requests by transmitting one or more of the rich transcripts that match the requests (page 4-5, System Architecture, server and browser).

However, ***Colbath*** does not disclose the broad audio classes including a phoneme-based vowel class, a phoneme-based fricative class, and a non-phoneme based bandwidth class. ***Liu*** discloses Audio classification logic configured to classify the input

audio data into at least one of a plurality of broad audio classes, the broad audio classes including a phoneme-based vowel class (Section 3 Phone-Class Decode, paragraphs 3 and 4), a phoneme-based fricative class (Section 3 Phone-Class Decode, paragraphs 3 and 4), and a non-phoneme based gender class (Section 3 Phone-Class Decode, paragraphs 3 and 4).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use the phone-based models of ***Liu*** in ***Colbath***, since it would result in improvements in speaker change detection accuracy, speech recognition accuracy and speed, as disclosed in ***Liu*** (Abstract).

Additionally, ***Stanford*** discloses a speech recognition system that enables recognition of high bandwidth or telephony (low bandwidth) speech signals (column 2 lines 30-32 and column 8 lines 36-44). ***Stanford*** states that low bandwidth speech reduces the accuracy of speech recognizers (column 3 lines 37-39), and discloses a system that trains and uses two separate codebook and phoneme models, one for low bandwidth speech and one for high bandwidth speech (column 8 lines 36-44). The addition of the low bandwidth recognition model improves recognition accuracy for low bandwidth input, such as telephone speech.

Therefore it would have been obvious to one or ordinary skill in the art at the time of the invention to have at least one broad audio class include a non-phoneme based bandwidth class in ***Colbath***, since one of ordinary skill in the art has good reason to

pursue the options within his or her technical grasp in order to accurately recognize and segment audio information for indexing, regardless of the input quality of the audio.

As per claim 24, this claim recites limitations similar to those recited in claim 8, and is therefore rejected for similar reasons.

As per claim 25, this claim recites limitations similar to those recited in claim 9, and is therefore rejected for similar reasons.

As per claim 26, this claim recites limitations similar to those recited in claim 6, and is therefore rejected for similar reasons.

As per claim 27, ***Colbath*** in view of ***Liu***, and further in view of ***Stanford*** disclose the method of claim 23, and ***Colbath*** further discloses wherein the indexer further includes at least one of a speaker clustering component, a speaker identification component, a name spotting component, and a topic classification component (page 2, Component Technologies).

As per claim 28, this claim recites limitations similar to those recited in claim 1, and is therefore rejected for similar reasons.

As per claims 29 and 30, these claims recite limitations similar to those recited in claims 4 and 5, and are therefore rejected for similar reasons.

As per claim 31, this claim recites limitations similar to those recited in claim 8, and is therefore rejected for similar reasons.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Dorothy Sarah Siedler whose telephone number is 571-270-1067. The examiner can normally be reached on Mon-Thur 9:30am-5:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DSS

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SUPERVISORY PATENT EXAMINER